Lecturer 4, 5

Pulse Code Modulation

Prepared by Prof

Mahmoud Ahmed Attia Ali

Department of Electronics and Communications
Faculty of Engineering
Tanta University
Egypt

TIP
For additional advice see
Dale Carnegie Training®
Presentation Guidelines

Contents

- Generation of PCM Signal.
- Quantization and Coding.
- Compading (Compression & Expansion):
  - µ Law.
  - A Law
- American and European Standard.
- Intersymbol Interference.

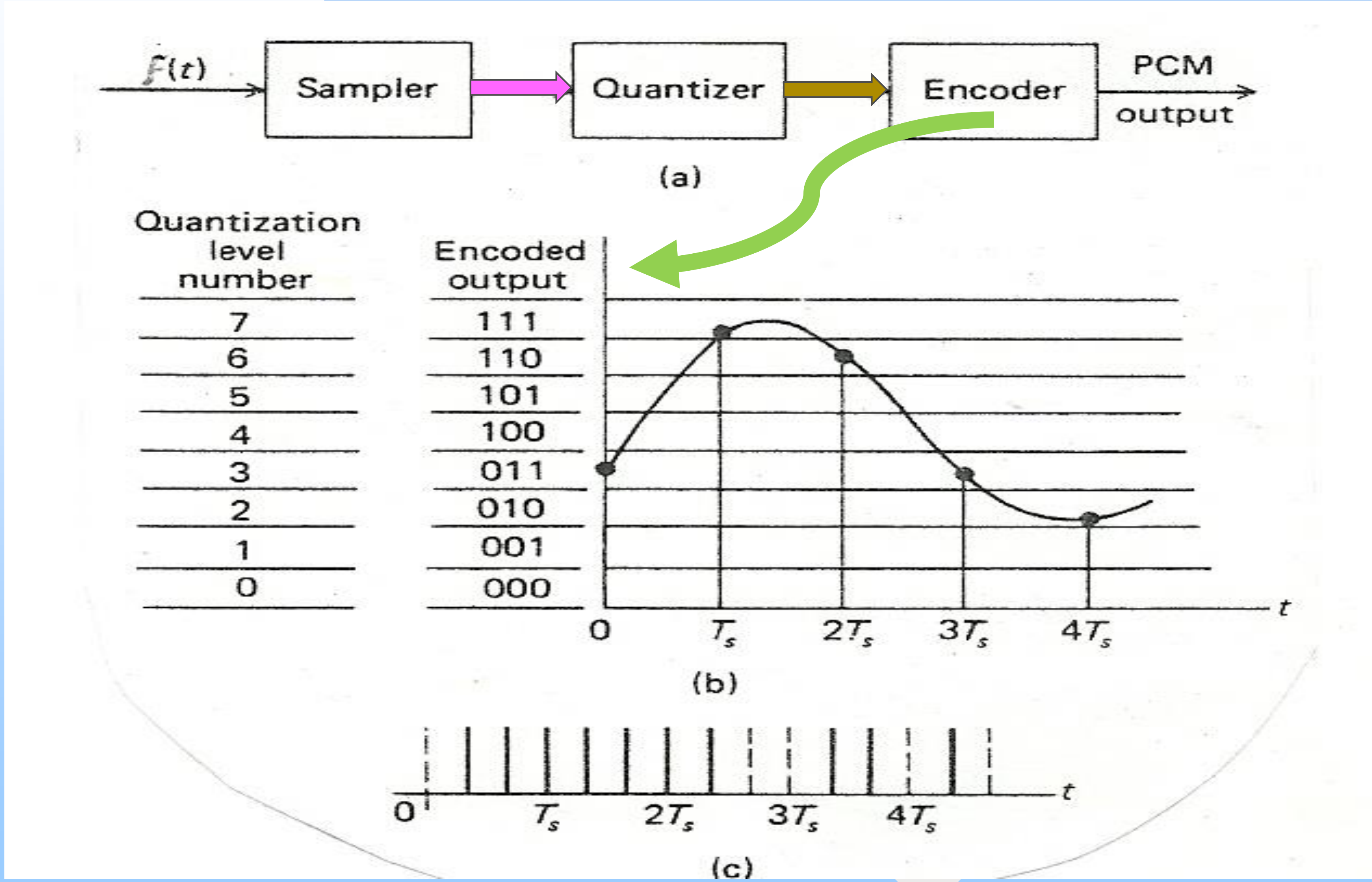
PCM

Generation of PCM

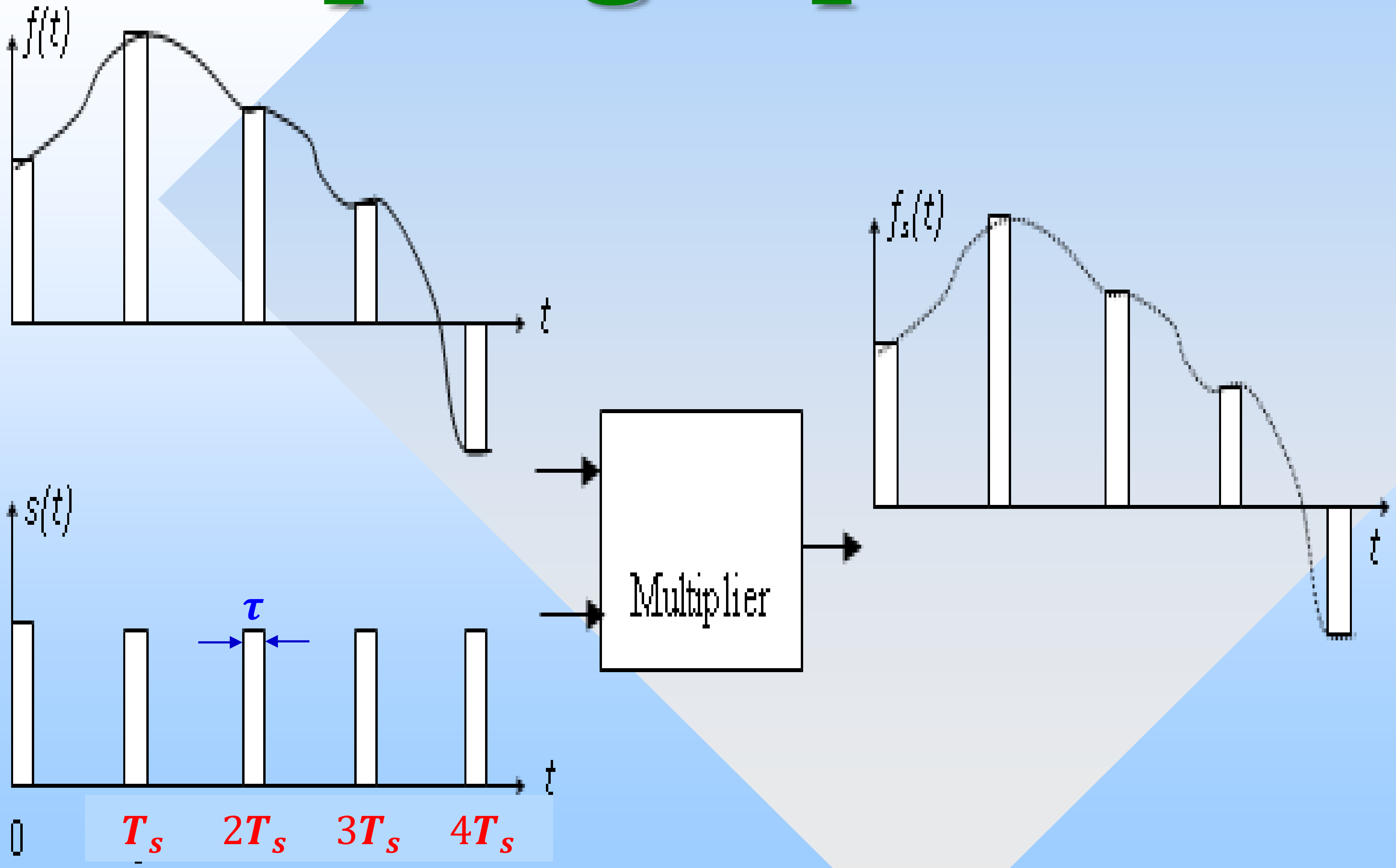
Consists of three processes:

- **Sampler:** Message signal $f(t)$ is first sampled by a rate $f_s > 2f_m$.
- **Quantizer:** Sample values are then quantized to a certain levels.
- **Encoder:** Quantization levels are encoded into binary sequence.

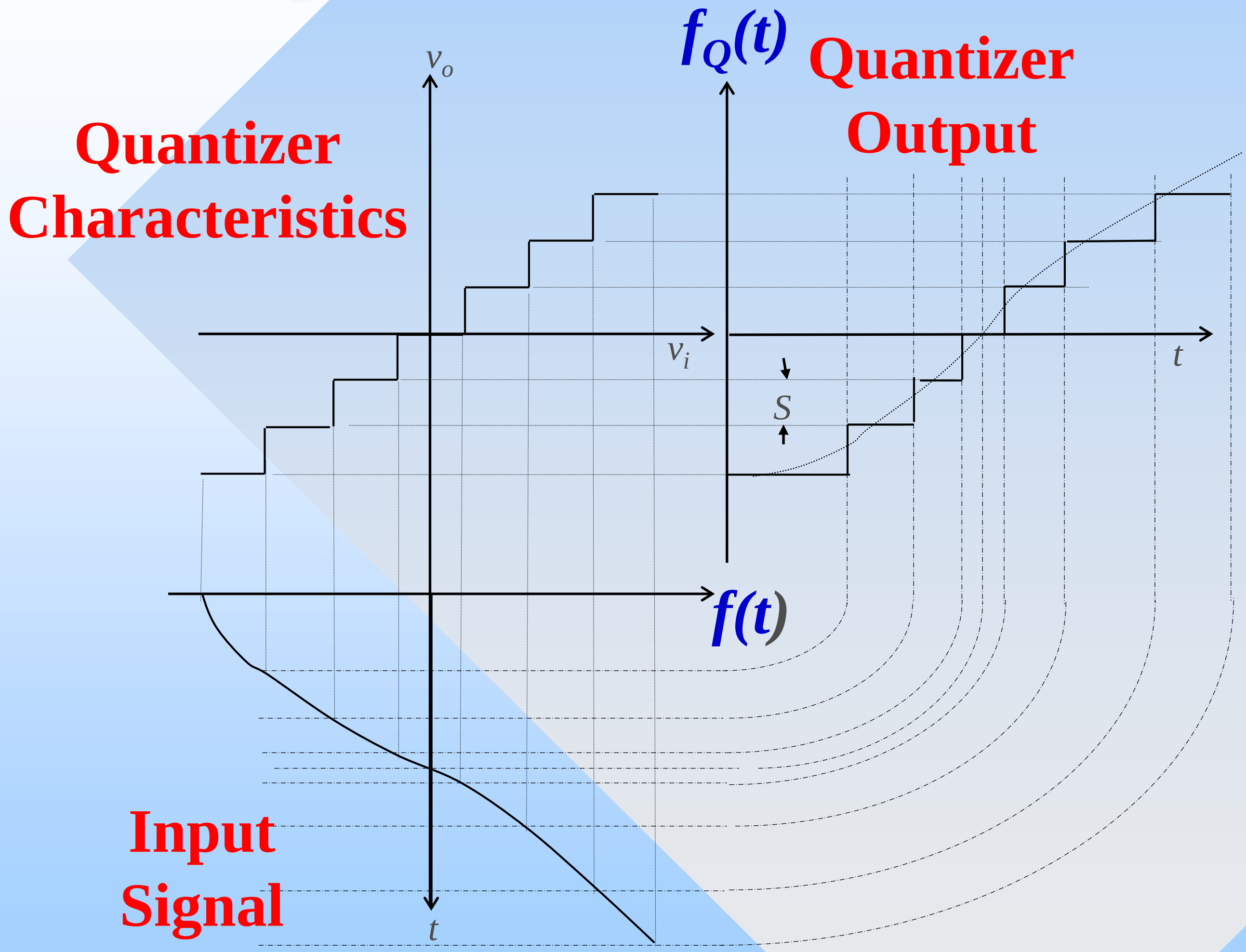
PCM Block Diagram



Sampling Operation



Quantization



Quantization Error

- Difference between original signal $f(t)$ and its quantized approximation $f_Q(t)$
- Why termed as quantization noise?
  - Affects the signal amplitude.
  - May be added to or subtracted from the signal.
  - Expected average value is zero.
- Maximum value is ½ least significant

$$Q_{maximunm} = \frac{v_{LSB}}{2}$$

- Upon reconstruction:
  - Added noise may be removed.
  - Sometimes errors take place.

Quality of Quantization

- Quality of approximation improved by reducing the step size.
- Tests for speech indicate that:
  - 2 levels are understandable but quite noisy.
  - 8 or 16 levels are sufficient for a good intelligibility.
  - 128 or 256 are usually used to ensure high quality.
- Tests for color TV:
  - 64 levels gives only good color TV.
  - 512 levels is used for commercial color TV.

Missed Signal Details

- In quantizing some details are lost.
- It is impossible to reconstruct the original.
- However, there is no need to transmit all signal details:
  - Ears and eyes in hearing and watching is limited:
    - Ear could not distinguish small distortion.
    - Eye has a limited resolution.
  - Also, due to noise, detector will not able to distinguish fine variation.

Probability of Level Error

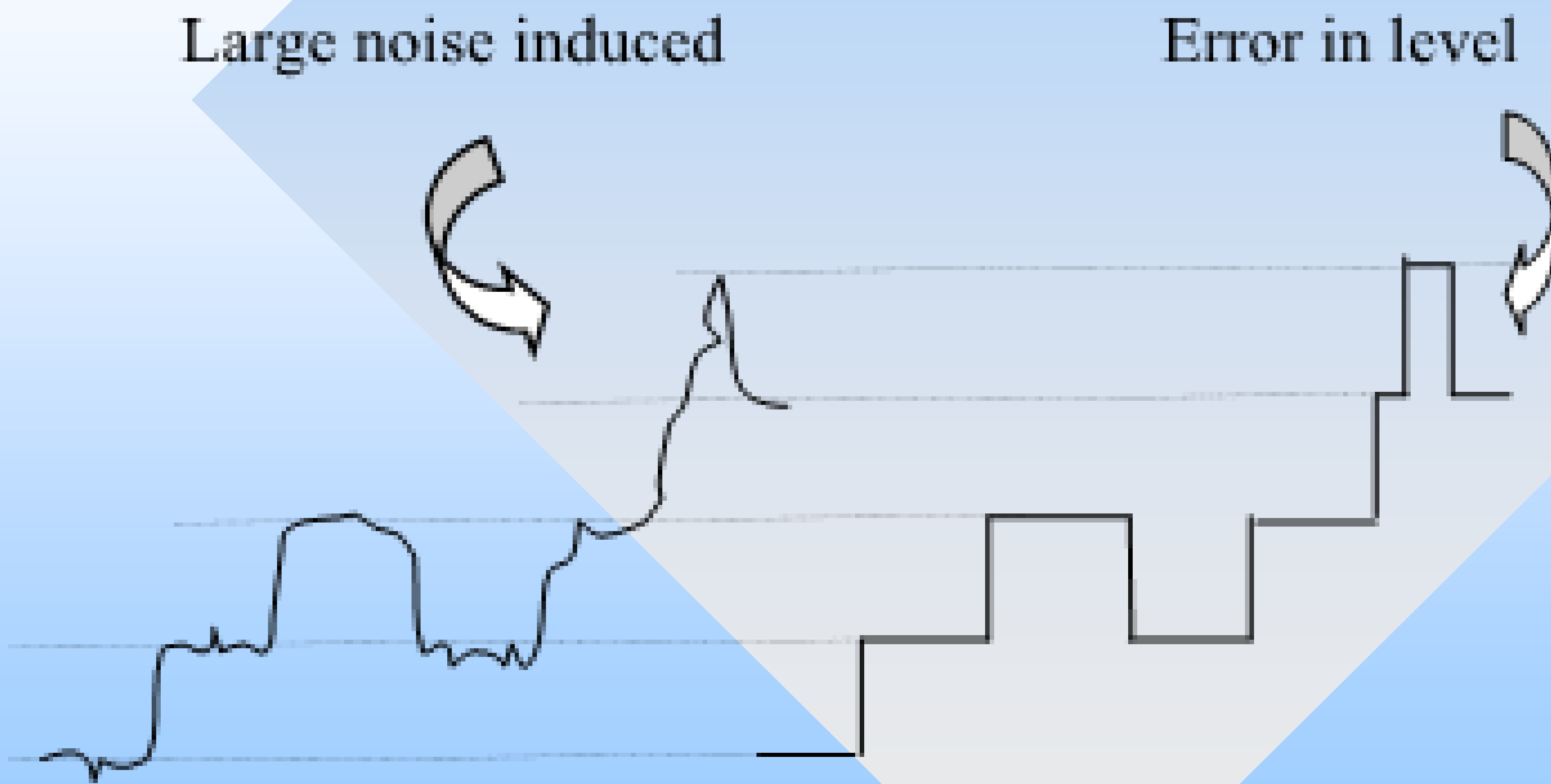


Fig.1.18: Noise Removing or Level Error

Encoding

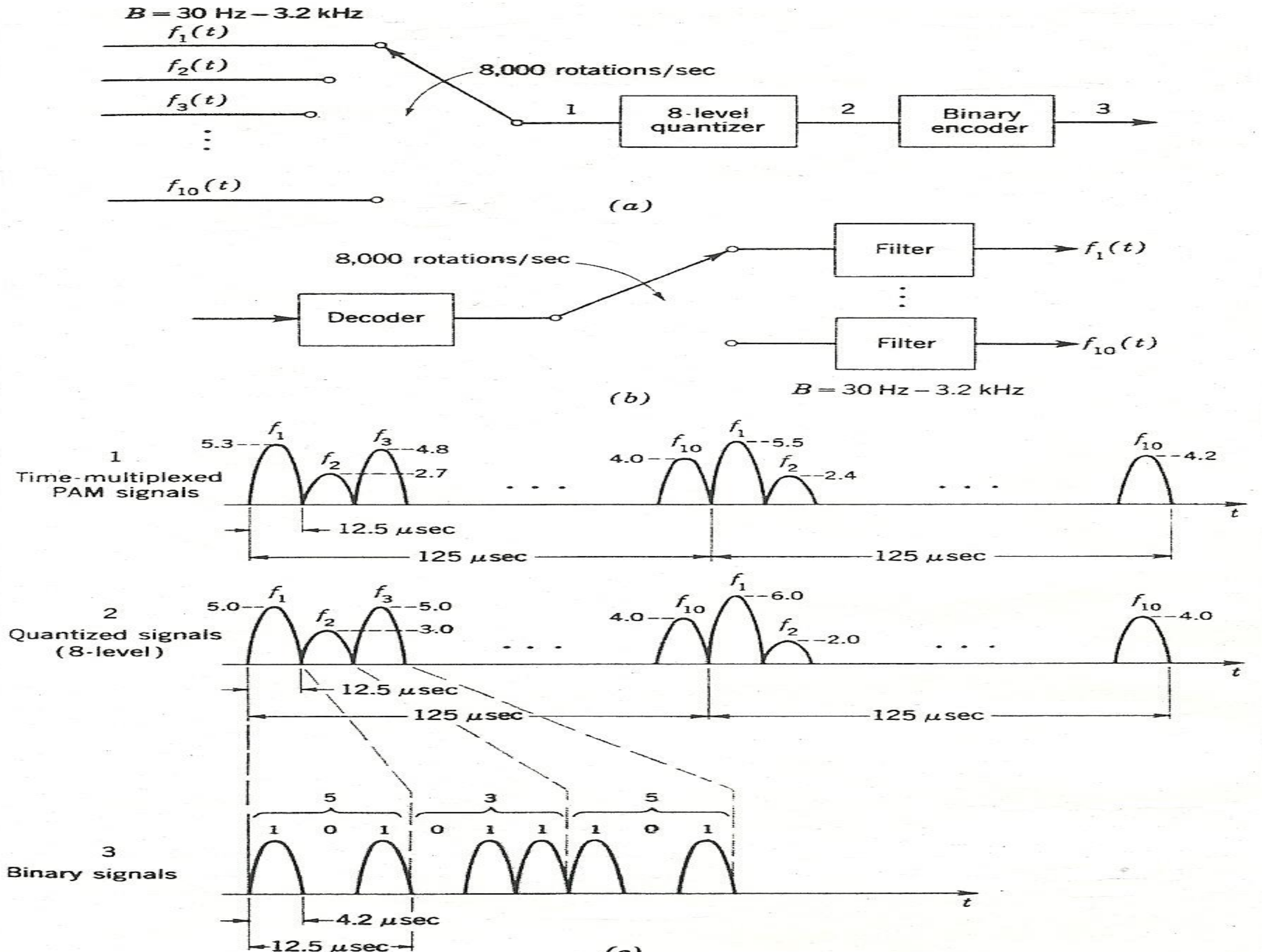
- A binary code where n equals 2.
- Number of quantization levels M is related to the number of bits per sample n as:

$$M = 2^n$$

- Generally, bandwidth for transmission of pulse train is inversely proportional to its width and depends on its shape.
- Therefore, bandwidth is roughly given by the reciprocal of the time slot.

Example

- **Assume 10 channels PCM System**
  - Sampled PAM,
  - Quantized, and
  - PCM using 8 levels quantizer.
- **Required Bandwidths:**
  - PAM signal bandwidth is 1/12.5 μsec = 80 kHz.
  - Quantized PAM signal bandwidth is 80 kHz.
  - PCM signal bandwidth is 1/ 4.2 μsec = 240 kHz.



B = 30 Hz – 3.2 kHz
f1(t)
f2(t)
f3(t)
f10(t)
8,000 rotations/sec
1
8-level quantizer
2
Binary encoder
3
(a)
8,000 rotations/sec
Decoder
Filter
f1(t)
Filter
f10(t)
B = 30 Hz – 3.2 kHz
(b)
1
Time-multiplexed PAM signals
5.3
4.8
2.7
4.0
5.5
2.4
4.2
12.5 μsec
125 μsec
125 μsec
2
Quantized signals (8-level)
5.0
5.0
3.0
4.0
6.0
2.0
4.0
12.5 μsec
125 μsec
125 μsec
5
3
5
1 0 1
0 1 1
1 0 1
3
Binary signals
4.2 μsec
12.5 μsec
t
(c)

Exercises

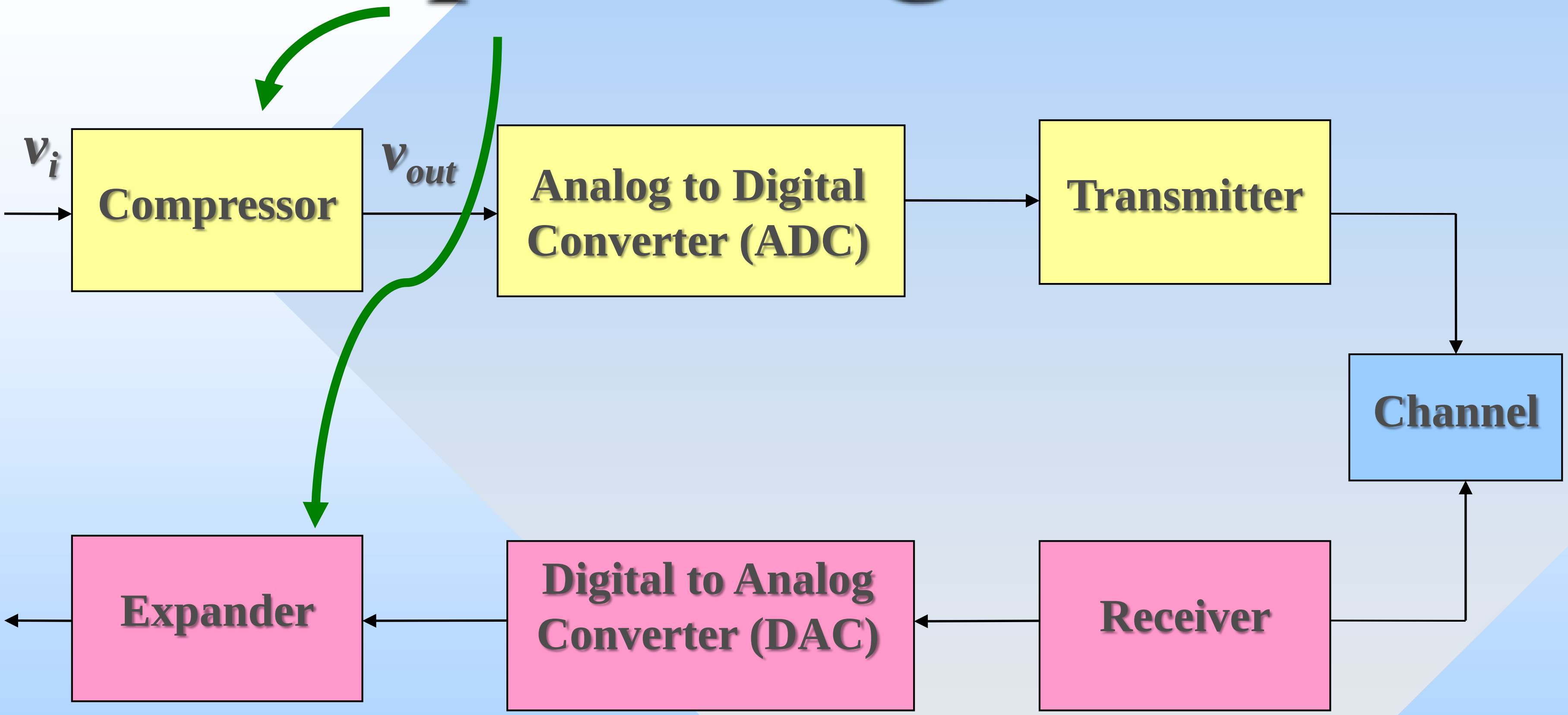
- Indicate the advantages of PCM systems when compared to PAM or Delta techniques.
- What is difference between the unipolar and the bipolar quantization?
- What is difference between mid-rise and mid-tread quantization processes?
- What is difference between the uniform and the nonuniform quantization?

Companding

Reason of Companding

- Small signals will have a poorer signal to quantization noise ratio SQR than large.
- So, it is better to have moderate levels for both low and high variations of the signal:
  - Increase low signal levels to more moderate levels so that SQR could be increased.
  - Reduce high signal levels to more moderate levels in order to decrease high SQR.

Companding Process



Companding Laws

- Companding means compression and expansion.
- Compression by using special designed diodes prior to sampling circuit
- Whereas expansion is attained with diodes after the receiver LPF.
- Voice signal require a constant SQR over a wide dynamic range DR. This requires a logarithmic compression ratio. There are two methods:
  - µ Law Companding.
  - A Law Companding.

μ Law Companding (USA and Japan)

μ Law Compression

$$v_o = \frac{\ln\left[\left\{1 + \mu\left(\frac{v_i}{V_{i,max}}\right)\right\}\right]}{\ln[1+\mu]} V_{o,max}$$

$V_{i,\text{max}}$: Max amplitude of input signal before compression.

$V_{o,\text{max}}$: Max amplitude of output signal after compression.

v_i: Amplitude of input signal before compression for $v_i > 0$

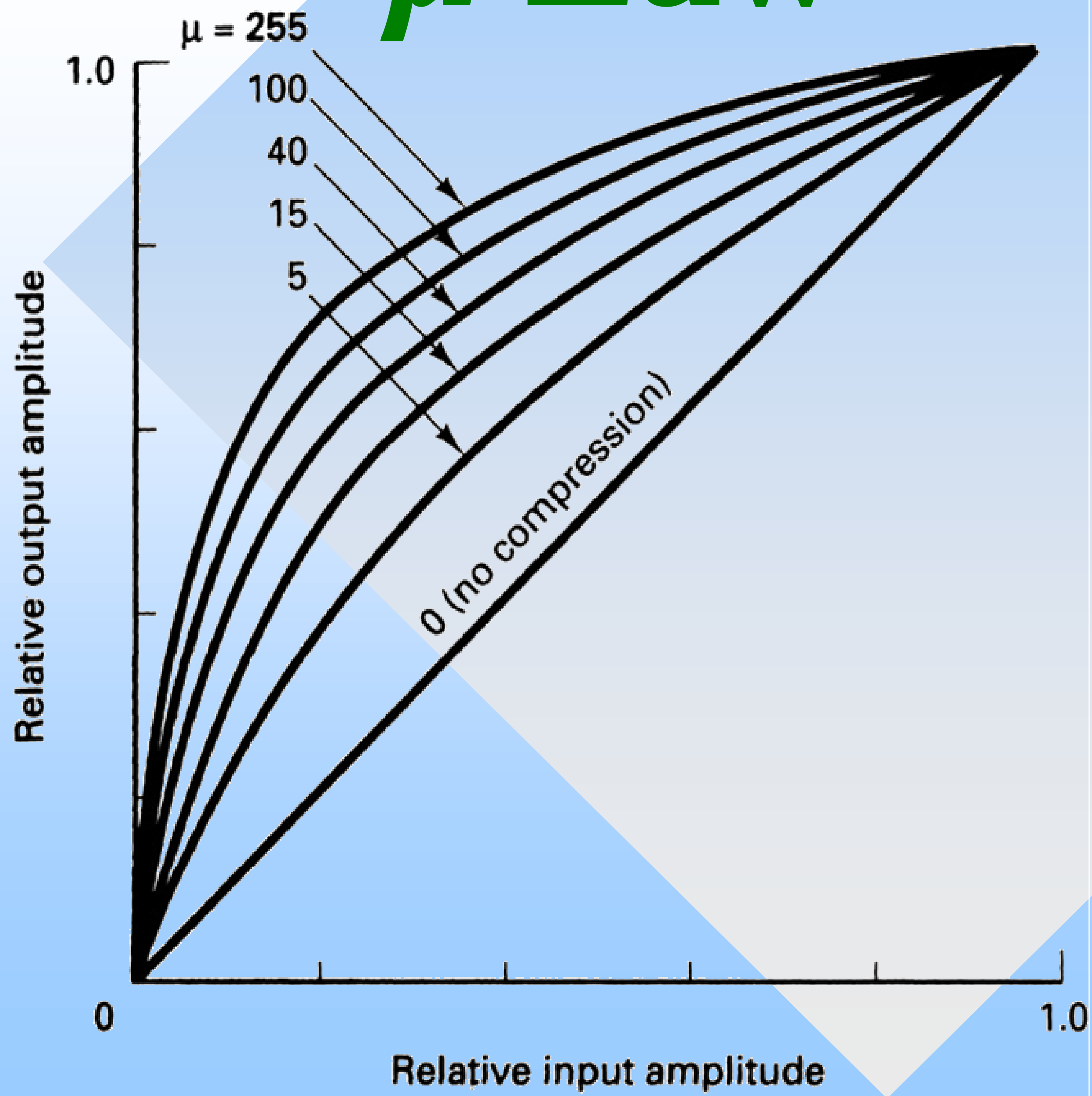
v_o: Amplitude of the output signal after compression.

μ: is a measure for the amount of compression.

μ: determines the range of signal power over which SQR is relatively constant.

- For DR = 40 dB, 7 bit PCM code uses μ = 100.
- For DR = 40 dB, 8 bit PCM code uses μ = 255.

μ Law



μ = 255
100
40
15
5
0 (no compression)
1.0
0
1.0
Relative output amplitude
Relative input amplitude

μ Law Expansion

On reception, received signal will be expanded to satisfy linearity. Henceforth, received signal is:

$$v_{o,r} = \frac{V_{i,r,max}}{\mu}\left[(1+\mu)\left(\frac{v_{i,r}}{V_{o,r,max}}\right) - 1\right] \;, v_{i,r} \geq 0$$

$V_{i,r,\mathrm{max}}$: Max amplitude of input received signal before expansion.

$V_{o,r,\mathrm{max}}$: Max amplitude of output received signal after expansion.

$v_{i,r}$: Amplitude of input received signal before expansion.

$v_{o,r}$: Amplitude of the output received signal after expansion.

A Law Companding (Europe CCITT)

A Law Compression

Compression characteristics is a true logarithm

$$v_o = \frac{\frac{Av_i}{V_{i,max}}}{1+\ln A} V_{o,max}, \quad 0 \le \frac{v_i}{V_{i,max}} \le \frac{1}{A}$$

$$v_o = \frac{1+\ln\left[\frac{Av_i}{V_{i,max}}\right]}{1+\ln A} V_{o,max}, \qquad \frac{1}{A} \le \frac{v_i}{V_{i,max}} \le 1$$

$V_{i,\max}$: Max amplitude of input signal before compression.

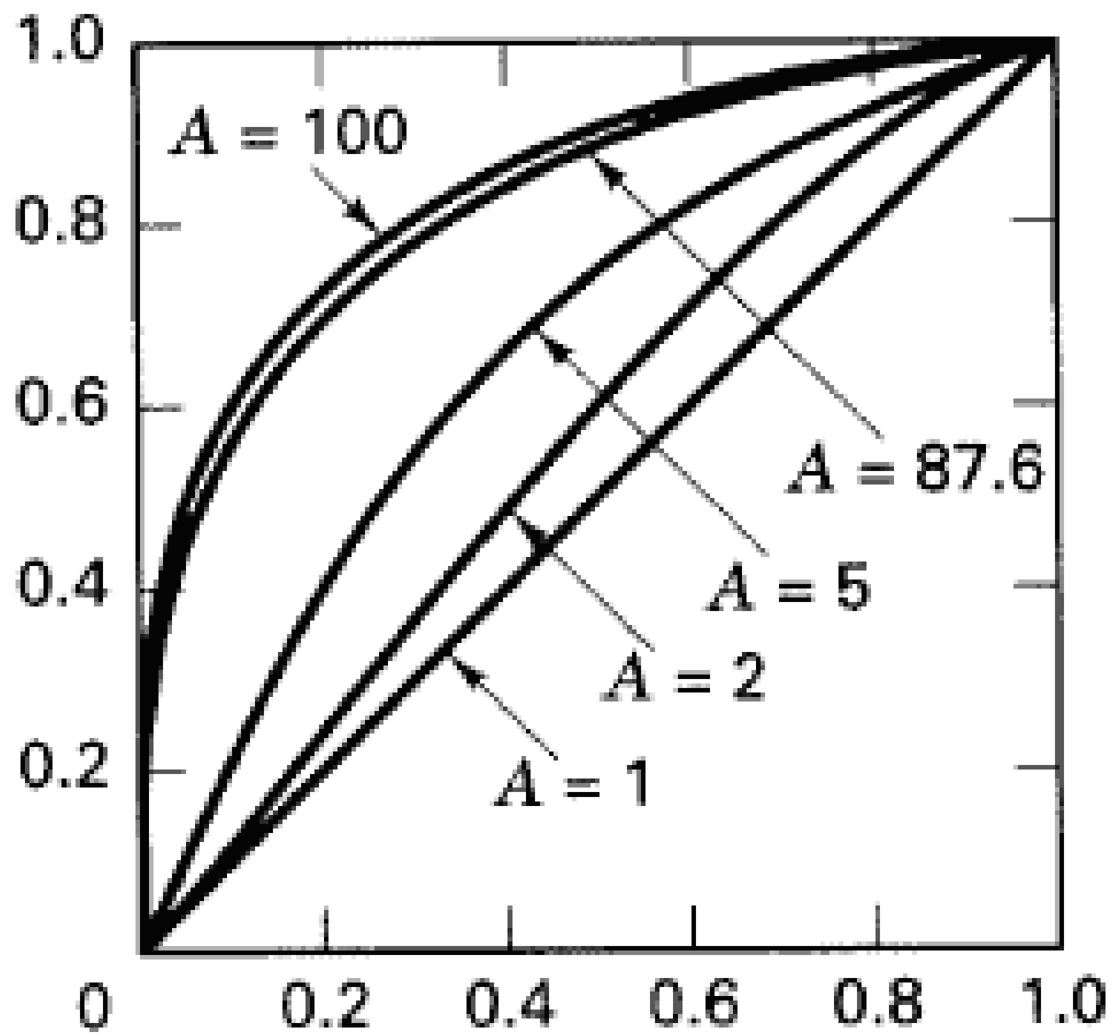
$V_{o,\max}$: Max amplitude of output signal after compression.

v_i: Amplitude of input signal before compression.

v_o: Amplitude of the output signal after compression.

Optimum value for voice transmission is $A = 87.6$.

A Law



Data Rate of PCM

- Speech signal for telephone has $f_m = 4$ kHz.
- So, the sampling rate is $2x4 = 8$ kHz, that is 8000 samples/sec.
- Sampling interval $1/8000 = 125\ \mu$ sec/Frame
- Each sample is encoded into 8 bits/sample.
- So, the data rate for PCM signal is:

$$R_{PCM} = (8000\ \text{samples/sec})(8\ \text{bits/sample})$$
$$= 64\ \text{k bits/sec}$$

North American PCM T1 Standard

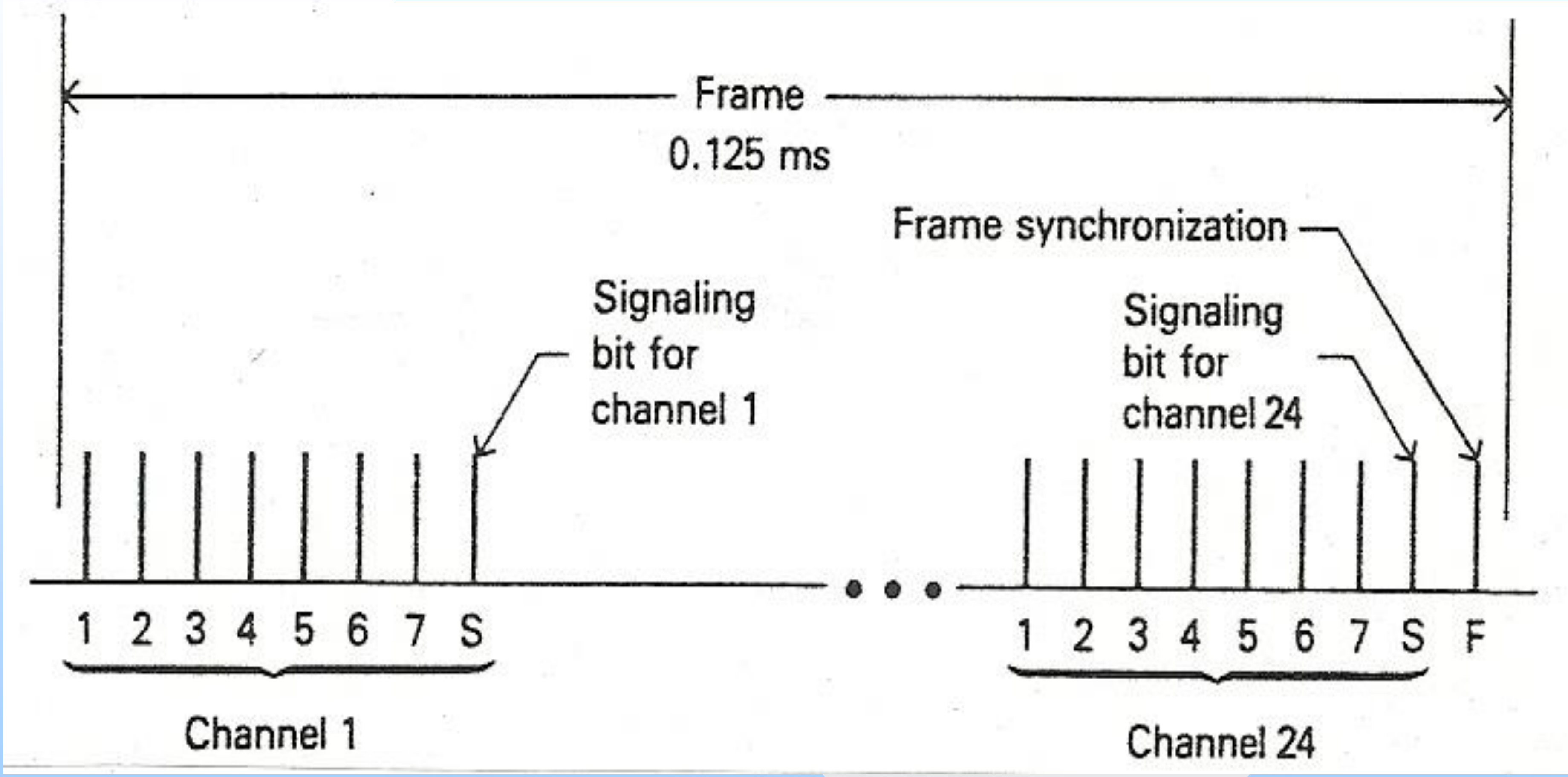
- Bell System in USA introduced 24 channel PCM in1960s for digital voice over short haul distances of 10 to 50 miles “T1”.
- T1 has found widespread adoption in US, Canada, and Japan.

PCM T1 Standard

- Early, it uses $2^7 = 128$ quantization levels.
  - Each sample is quantized into 7 bits.
  - 1 bit for establishing calls (signaling).
- Recently, $2^8 = 256$ levels have been adopted for quieter system with less distortion.
- 24 channels are time multiplexed, sampled, and coded into 8 bit PCM formats in addition to 1 bit for frame synchronization.
- The frame consists of $24 \times 8 + 1 = 193$ bits.

$$R_{T1} = (193\ \text{bits/Frame})(1\text{Frame}/125\mu\text{sec}) = 1.544\ \text{Mbps}$$

North American PCM Standard for Short-Haul Telephone [T1 System]



Intersymbol Interference ‘ISI’

- First; bandwidth is restricted, so transmitted waveforms rounded off, reshaped or distorted in a way analogous to crosstalk in PAM.

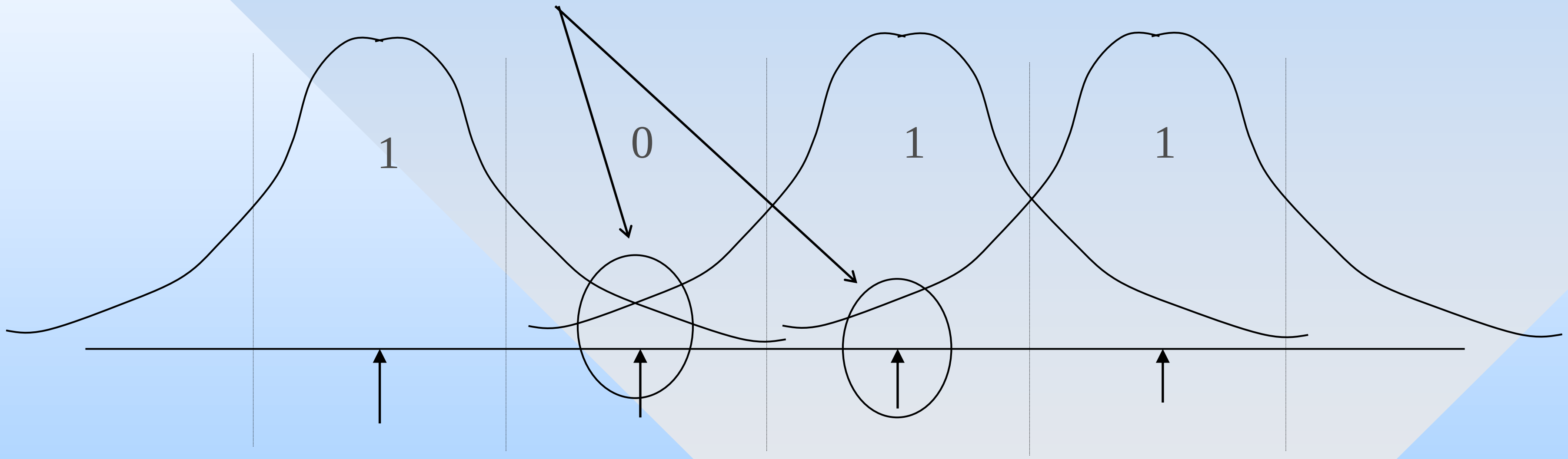
- Second; where digital pulses will modulate a carrier for transmission over long distances, pulse shaping is a must.

Crosstalk and ISI

- In un-quantized PAM, adjacent time slots are often associated with different message channels, and the term crosstalk is appropriate.
- In PCM adjacent bits are more generally symbols in the code representation of a single quantized sample, hence the term inter-symbol interference.

ISI

Inter-symbol Interference



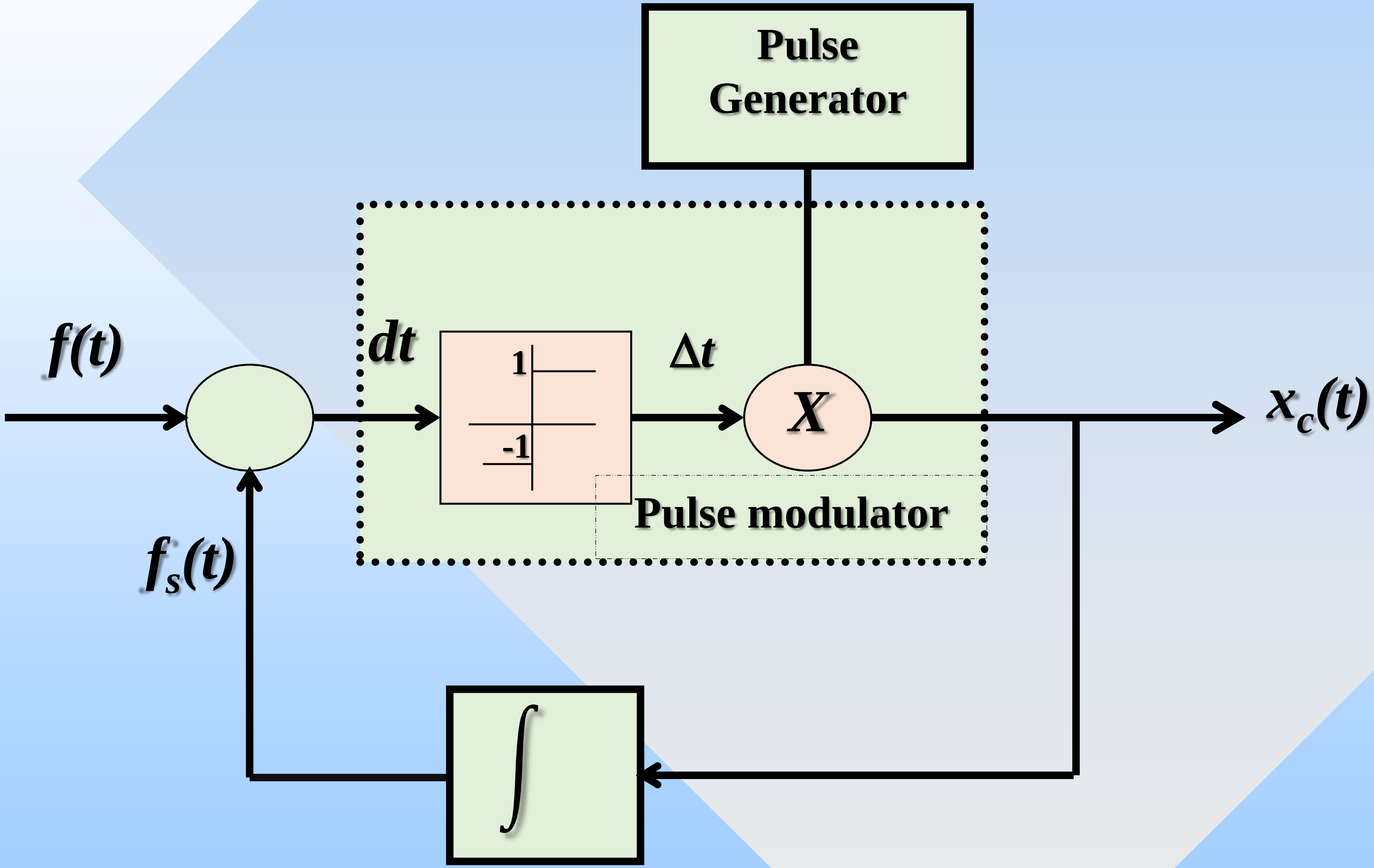
DM Transmitter

- $f_s(t)$ is a stair step approximation of $f(t)$.
- Modulator input is: $d(t) = f(t) - f_s(t)$
- $d(t)$ is hard-limited and multiplied by the pulse generator to yield the output:

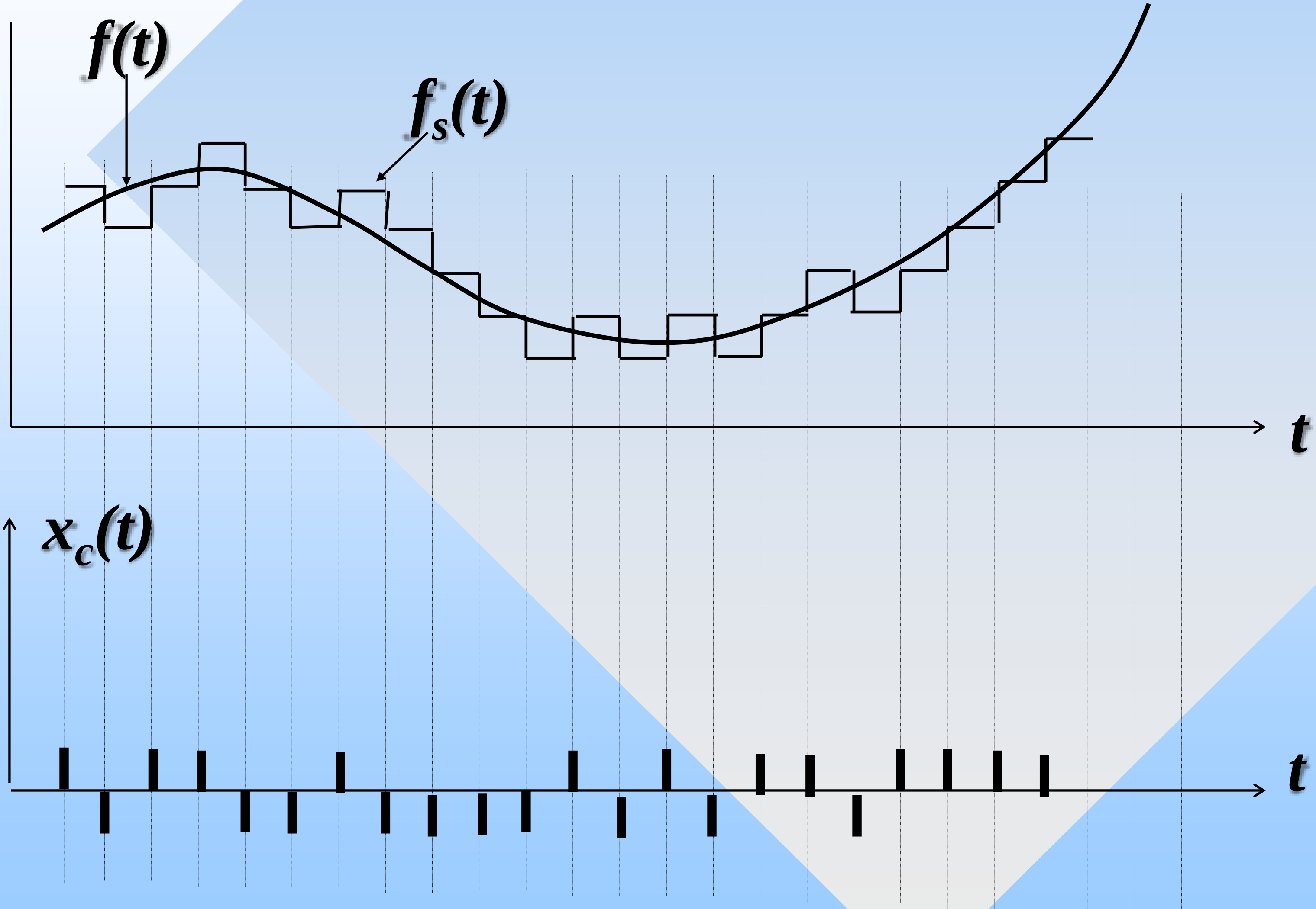
$$x_c(t) = \Delta(t) \sum_{-\infty}^{\infty} \delta(t - nT_s) = \sum_{-\infty}^{\infty} \Delta(nT_s)\delta(t - nT_s)$$

$$f_s(t) = \int_0^t x_c(t)\,dt = \sum_{n-\infty}^{\infty} \Delta(nT_s) \int_0^t \delta(y - nT_s)\,dy$$

DM Transmitter



Delta Modulation Waveforms

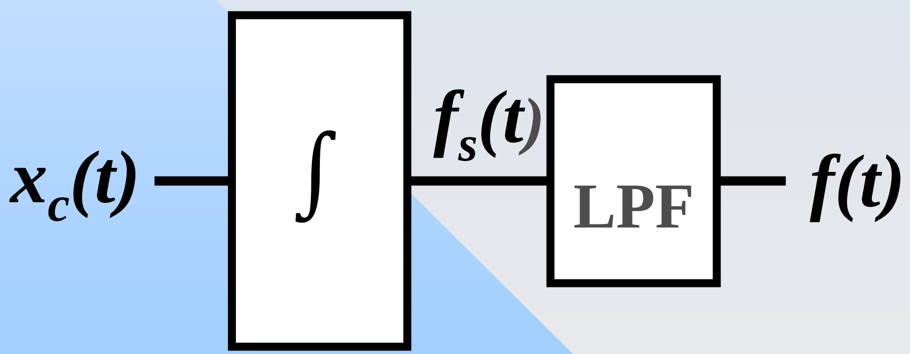


DM Receiver

Demodulation of DM is simple by:

- Integrating $x_c(t)$ to form stair approximation $f_s(t)$
- That is low pass filtered to suppress discrete jumps.

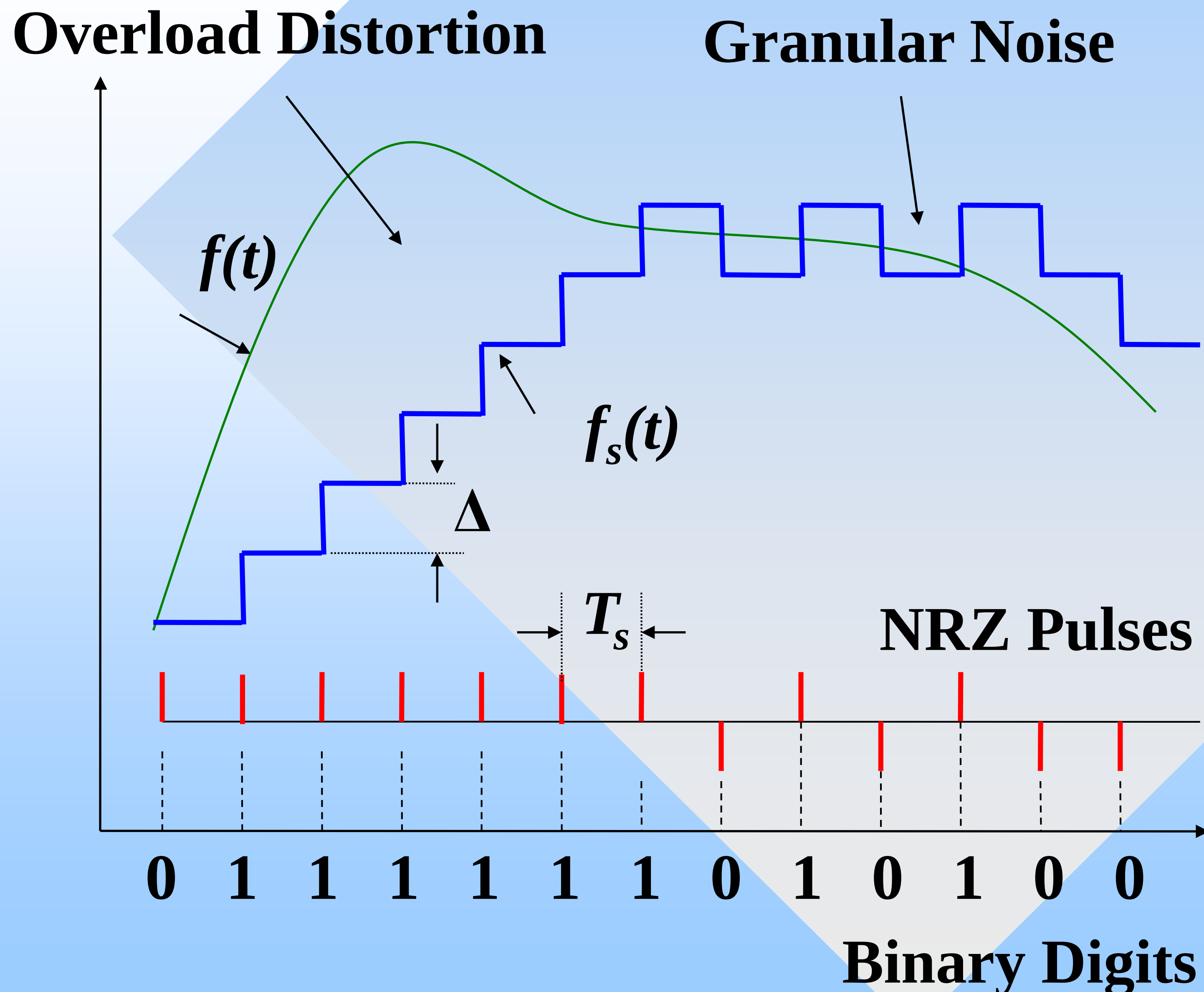
Since LPF approximates an integrator, it is often possible to eliminate the integrator.



Granular Noise

When changes in $f(t)$ are less than the step size, DM no longer follows the signal and it produces a train of alternating positive and negative pulses:

- Similar to quantization noise in conventional PCM.
- When $f(t)$ has constant amplitude reconstructed signal has variations that were not present in the original signal.
- It can be reduced by decreasing the step size (**high resolution**).


Overload Distortion
Granular Noise
f(t)
fs(t)
Δ
Ts
NRZ Pulses
0 1 1 1 1 1 1 0 1 0 1 0 0
Binary Digits

Slope Overload Distortion

When the analogue signal changes at a faster rate [i.e, $f(t)$ slope is too high].

Its slope is greater than DM can maintain.

Can be reduced by:

- Increasing the clock frequency.
- Increasing the step size (**Low resolution**)

Adaptive DM Transmitter

If $f(t)$ is relatively constant, the pulses $x_c(t)$ will alternate sign.

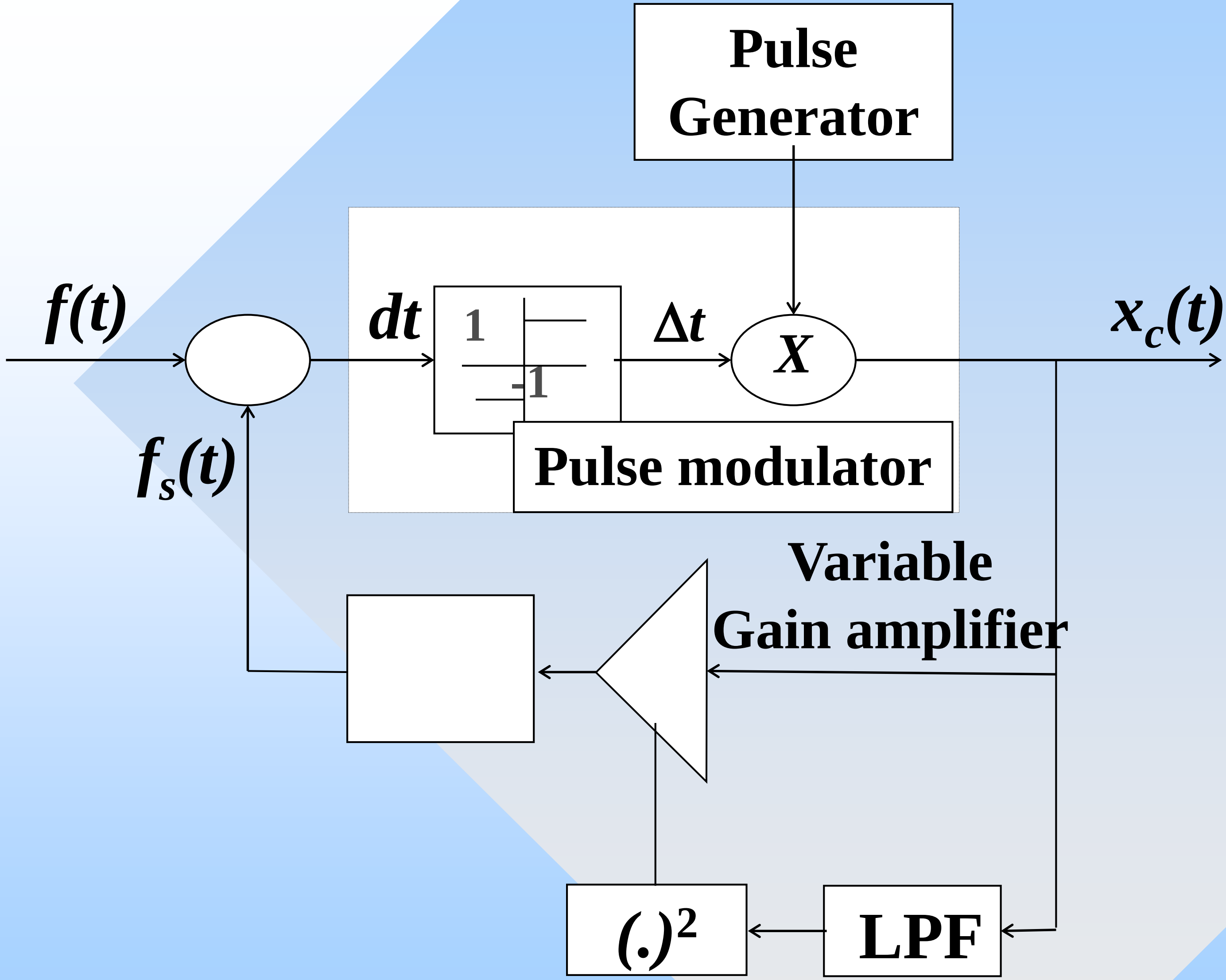
So the dc value of LPF output is nearly zero

(**minimum step size**)

If $f(t)$ increases or decreases rapidly, $x_c(t)$ will have same polarity over that period.

So the dc value of LPF output is large

(**larger step size**)



Adaptive Delta Modulation

Adaptive DM Receiver

The receiver of ADM should be adaptive also